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SPECIFICATION

TITLE OF THE INVENTION

Coding Apparatus for Digital Signal

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BACKGROUND OF THE INVENTION

Field of the Invention

This invention relates to a coding apparatus adapted for implementing the so-called block floating processing to an input digital signal, and to transform a signal on the time base to a signal on the frequency base to divide it into signal components in a plurality of critical bands to carry out bit allocation of the signal components every respective critical bands, thus to encode them.

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Description of the Prior Art

As one of technologies for implementing bit compression to an audio signal, etc. to encode it, there is known so called a block floating technology to divide input data into data blocks every a predetermined number of words to carry out a floating processing every respective blocks. In accordance with this block floating technology, an approach is employed to seek or search for the maximum one (maximum absolute value) of absolute values of respective words in a block to carry out a floating processing by using this maximum absolute value as a common floating coefficient for all the words in the corresponding block.

Further, there is also known an orthogonal transform coding to transform (so called orthogonally transform) a signal on the time base to a signal on the frequency base to encode it. As this orthogonal transform coding technology, there is such a technology to divide, e.g., audio PCM data into data blocks every a predetermined number of words to carry out a Discrete Cosine Transform (DCT) processing every respective blocks. In addition, there is also known such a technology to divide a signal on the frequency base into signal components every so called critical bands to apply an adaptive bit allocation thereto every respective bands. The number of bits allocated is determined in dependency upon allowed noise levels every critical bands in which the so-called masking is taken into consideration.

Meanwhile, there are many instances where the operational processing for such an orthogonal transformation is executed by using a FIR (Finite-duration Impulse-Response) filter of the multi-tap type. At this time, such an operational processing includes a coefficient multiplication processing and/or an operation for taking a sum total, etc., resulting in the possibility that the number of digits may be increased so that it overflows. To prevent such an overflow, it is required to allow for in advance the number of digits for operation by, e.g., several orders of digits greater than the number of digits of input data. For such a multi-digit operation, a high performance DSP (Digital Signal Processing unit) is required, and it takes

much time as well. Accordingly, a more simple orthogonal transform processing is expected.

In view of this, a technique has been proposed to implement the above-mentioned block floating processing to input data prior to the orthogonal transform processing to carry out bit compression thereof to thereby reduce in advance the number of digits of data subject to the orthogonal transform operation.

Further, a technique has been also proposed to adaptively vary the length of a block for the orthogonal transform processing in dependency upon an input signal. Such a technique is employed because particularly in the case of dividing in advance an input signal into signal components in several (e.g., about three) bands to carry out orthogonal transform processing thereof every respective bands, a method of varying the length of a block in the magnitude of changes in time or a pattern, etc. of signals in respective bands permits a more efficient coding than a coding method in which the length of a block is fixed.

It is to be noted that in the case where the above-mentioned technology for implementing a block floating prior to the orthogonal transform processing and the above-mentioned technology for adaptively changing the length of a block in dependency upon an input signal are combined to carry out coding, respective processing are independently required, resulting in the drawback that a quantity subject to processing is increased.

In actual terms, for example, as shown in Fig. 15, a

relatively large block BL is divided in advance into several blocks (e.g., four blocks), thus to prepare small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4}. As indicated by step S31 of Fig. 16, respective energies of these small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4} are calculated for determination of a size of a variable length block (block length) to determine, at the next step S32, a block size in dependency upon energies of the respective blocks. Then, at step S33, a maximum absolute value within a determined block is calculated to implement a block floating processing on the basis of the maximum absolute value thus calculated. At the next step S34, the orthogonal transform processing such as DCT for this block is carried out.

In such a processing procedure, calculation of energies every respective small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4} for determination of block size (block length), and calculation of maximum absolute values of respective blocks for the block floating processing are required. As a result, a quantity subject to processing or the number of steps in processing by the so-called microprogram is increased.

Meanwhile, with respect to allowed noise levels determined every respective critical bands in consideration of the above-mentioned masking, a method has been proposed to correct such allowed noise levels by taking into consideration the so-called minimum audible characteristic in the hearing sense of the human being. In accordance with this method, an allowed

noise level already calculated and a minimum audible level are compared with each other to assume a greater one as a new allowed noise level.

The allowed noise level in which the masking is taken into consideration is considered to be the same level in the above-mentioned critical band. However, since a measured value of the minimum audible limit is given by using a sine wave, particularly in the region where the critical bandwidth is broad as in a high frequency band, a value at a low frequency portion and a value at a high frequency portion within the same critical band differ from each other ~~to much~~ degree. For this reason, if an approach is employed to give a single minimum audible limit level every critical band, an error would become large, resulting in the possibility that there may take place useless number of allocated bits at the high frequency portion within the critical band.

In addition, although it is a conceivable to finely divide the critical band into small bands to give minimum audible limits every small divided bands, this is not preferable in that a quantity of information to be transmitted is increased.

Summary of the Invention

With such actual circumstances in view, this invention has been proposed, and its object is to provide a coding apparatus for a digital signal wherein in such cases of implementing a

block floating prior to the orthogonal transform processing and adaptively changing the length of a block in dependency upon an input signal, the apparatus is constructed so that it can reduce a quantity subject to processing.

Another object of this invention is to provide a coding apparatus for a digital signal wherein in the case of dividing an input signal on the frequency base into signal components in critical bands to respectively apply adaptive bit allocation thereto on the basis of allowed noise levels, the apparatus is constructed so that it can reduce an error of a minimum audible level within a critical band ~~when it~~ is employed as the allowed noise level.

To achieve the above-mentioned objects, according to a first aspect of this invention, there is provided a coding apparatus for a digital signal comprising: a band division filter for dividing an input digital signal into signal components in a plurality of frequency bands; a block floating circuit for implementing, every block, floating processing to an output signal of the band division filter; a plurality of orthogonal transform circuits for orthogonally transforming respective output signals on the time base of the block floating circuit to signals on the frequency base; and an adaptive bit allocation encoder for dividing output signals of the orthogonal transform circuits into signal components in critical bands to adaptively allocate bit numbers thereto on the basis of allowed noise levels

every respective critical bands, wherein the length in the time base direction of the block is caused to be variable, and the length in the time base direction of the block and floating coefficients at the time of the floating processing are determined on the basis of the same index.

The above-mentioned index may be given by a logical sum of absolute values of respective words.

Further, according to a second aspect of this invention, a coding apparatus for a digital signal featured by the first aspect of the invention may further comprise allowed noise level calculation means for calculating allowed noise levels obtained within the critical band every critical band; and comparison means for comparing the allowed noise level with a minimum audible level to raise or set a flag when the minimum audible level is higher than the allowed noise level, wherein, in the critical band where the comparison raises the flag, the level of the minimum audible curve is selected as the allowed noise level.

In the coding apparatus of the second aspect, the allowed noise level calculation means may be constructed to calculate an allowed noise level from an energy every critical band and a minimum audible curve, etc., and to further calculate an allowed noise level on the basis of an error between an output information quantity and a bit rate target value of the final coded data. Such an approach may be employed to increase or decrease allocated bits of respective unit blocks by using an

output of the allowed noise level calculation means.

Furthermore, the feature of the first aspect may be combined with the feature of the second aspect. Namely, in the above-mentioned configuration of the coding apparatus of the second aspect, the length in a time direction of the block may be caused to be variable, and the length in the time direction of the block and floating coefficients at the time of the floating processing may be determined on the same index.

In addition, the above-mentioned orthogonal transform circuit may be constructed as a Discrete Cosine Transform (DCT) circuit.

As taught by the above-mentioned aspects of this invention, the coding apparatus for digital signal of the invention is adapted to carry out a block floating processing of an input digital signal by a variable length block thereafter to implement an orthogonal transform processing thereto. In this coding apparatus, by determining the length of a variable length block and a floating coefficient of the block floating on the basis of the same index, thereby making it possible to reduce a quantity subject to quantization or the number of steps of a program.

Further, in accordance with the coding apparatus for digital signal featured above, when an allowed noise level every critical bands is determined by the minimum audible level, bit allocation is carried out by allowed noise levels every small bands obtained by further dividing the critical band to only transmit a flag

indicating this, thus to avoid the necessity of sending allowed noise levels every small bands. Accordingly, accurate allowed noise levels can be provided without increasing auxiliary information quantity. This leads to the fact that signal quantity can be improved without degrading bit compression efficiency. In addition, even if an absolute value of the minimum audible limit level is altered later, compatibility can be maintained.

Brief Description of the Drawings

Fig. 1 is a circuit diagram showing, in a block form, the outline of the configuration of a coding apparatus for a digital signal according to an embodiment of this invention.

Fig. 2 is a view showing an actual example of divided bands and formation of blocks in the time base direction in the respective bands in the embodiment.

Fig. 3 is a flowchart for explaining the essential part of an encoding operation in the embodiment.

Fig. 4 is a view showing a critical band used for explaining an encoding operation in the embodiment.

Fig. 5 is a flowchart for explaining the essential part of a decoding operation in the embodiment.

Fig. 6 is a view showing a critical band used for explaining a decoding operation in the embodiment.

Fig. 7 is a view showing the example where a block size in

the time base direction in respective bands is switched between two sizes in the apparatus of Fig. 1.

Fig. 8 is a view showing the example where a block size in the time base direction in respective bands is switched between three sizes in the apparatus of Fig.1.

Fig. 9 is a flowchart for explaining the operation of the embodiment.

Fig. 10 is a circuit diagram showing, in a block form, an actual example of allowed noise calculation circuit 20 of the apparatus shown in Fig. 1.

Fig. 11 is a view showing a bark spectrum.

Fig. 12 is a view showing a masking spectrum.

Fig. 13 is a view in which a minimum audible curve and a masking spectrum are synthesized.

Fig. 14 is a block diagram showing an actual example of a decoder to which the embodiment of this invention can be applied.

Fig. 15 is a view showing an example of the length of a block by the processing procedure in the prior art.

Fig. 16 is a flowchart showing an example of the procedure of a conventional block floating processing.

DESCRIPTION OF THE PREFERRED EMBODIMENT

A preferred embodiment of this invention will now be described with reference to the attached drawings.

A coding apparatus for a digital signal to which this

invention can be applied is directed to coding apparatuses for a digital signal, which are adapted for carrying out an efficient coding of an input digital signal such as an audio PCM signal, etc. by using respective technologies of subband division coding (SBC), adaptive transform coding (ATC), and adaptive bit allocation (APC-AC). In actual terms, in the apparatus of the embodiment shown in Fig. 1, a technique is employed to divide an input digital signal into signal components in a plurality of frequency bands, and to make a setting such that the bandwidths become broader according as ~~the~~ frequency shifts to a higher frequency band side to carry out orthogonal transform processing every respective frequency band~~s~~ ~~to~~ apply adaptive bit allocation to spectrum data on the frequency base thus obtained every so called critical band in which the hearing sense characteristic of the human being is taken into consideration which will be described later, thus to encode them. In addition, in the embodiment of this invention, a technique is also employed to adaptively vary a block size (block length) in dependency upon an input signal prior to the orthogonal transform processing, and to carry out a floating processing every block.

Namely, in FIG. 1, input terminal 10 is supplied with an audio PCM signal of, e.g., 0 to 20 KHz. This input signal is divided into a signal in the frequency band of 0 to 10 KHz and a signal in the frequency band of 10 K to 20 KHz by using a band division filter 11, e.g., the so-called QMF filter, etc., and the

signal in the frequency band of 0 to 10 KHz is further divided into a signal in the frequency band of 0 to 5 KHz and a signal in the frequency band of 5 k to 10 KHz similarly by using a band division filter 12, e.g., the so-called QMF filter, etc. The signal in the frequency band of 10 K to 20 KHz from the band division filter 11 is sent to a Discrete Cosine Transform (DCT) circuit 13 serving as an orthogonal transform circuit, the signal in the frequency band of 5 K to 10 KHz from the band division filter 12 is sent to a DCT circuit 14, and the signal in the frequency band of 0 to 5 KHz from the band division filter 12 is sent to a DCT circuit 15. Thus, these signals are subjected to DCT processing, respectively.

Meanwhile, in the embodiment of this invention, in order to reduce a quantity of operation in the orthogonal transform processing, a technique is employed to implement block floating processing to an input data on the time base which has not yet been subjected to the orthogonal transform processing to carry out bit compression thereof to release the above-mentioned block floating after that data has been subjected to the orthogonal transform processing.

Namely, in Fig. 1, data on the time base of respective bands obtained from the band division filters 11 and 12 are delivered to a block floating processing circuit 16, at which a block floating processing is carried out with respective blocks BL as shown in Fig. 15 being as a unit. At respective orthogonal

transform circuits (DCT, i.e., Discrete Cosine Transform circuits in the example of Fig. 1) 13, 14 and 15, as operation for the orthogonal transform processing is implemented to the data which have undergone such block floating processing. Thereafter release the above-mentioned block floating is released by a floating release circuit 17. In releasing the block floating, floating information from the block floating processing circuit 16 is used. Also in the case of determining floating coefficients in such a block floating processing, an approach may be adopted to take a logical sum of absolute values of respective words in a block as previously described.

An actual example of a standard input signal with respective to blocks every respective frequency bands delivered to the DCT circuits 13, 14 and 15 is shown in FIG. 2. In the actual example of FIG. 2, a scheme is employed such that according as the frequency shifts to a higher frequency band side, the frequency bandwidth is caused to be broad and the time resolution is caused to be high (the block length is caused to be short). Namely, for a signal in the frequency band of 0 to 5 KHz on a lower frequency band side, one block BL_L is caused to have, e.g., 1024 samples. For a signal in the medium frequency band of 5 K to 10 KHz, that signal is divided into signal components in blocks BL_{M1} and BL_{M2} each having a length $T_{BL}/2$ one half of a length T_{BL} of the block BL_L on the low frequency band side. For a signal in the frequency band of 10 K to 20 KHz on a higher frequency band side,

that signal is divided into signal components in blocks BL_{H1}, BL_{H2}, BL_{H3} and BL_{H4} each having a length $T_{BL}/4$ one fourth of that of the block BL_L on the lower frequency band side. It is to be noted that in the case where the frequency band of 0 to 22 KHz is taken into consideration for an input signal, the low frequency band is in a range from 0 to 5.5 KHz, the medium frequency band is in a range from 5.5 K to 11 KHz, and the high frequency band is in a range from 11 K to 22 KHz.

It is to be noted that, in the embodiment of this invention, as described later, the block size (block length) is caused to be variable in dependency upon an input signal, and determination of the block size is carried out on the basis of a maximum absolute value used also for determining floating coefficients of the block floating.

Turning back to FIG. 1, spectrum data or DCT coefficient data on the frequency base, obtained as the result of the DCT processing at the respective DCT circuits 13, 14 and 15, are subjected to releasing of the block floating processing at the floating release circuit 17, and are then combined every the so-called critical band. The data thus obtained are further sent to an adaptive bit allocation encoder 18. This critical band is a frequency band divided in consideration of the hearing sense characteristic of the human being, and is a band that a narrow band noise having the same intensity as that of a pure sound in the vicinity of a frequency thereof has when the pure sound is

masked by that noise. This critical band is such that according as the frequency shifts to a higher frequency band side, the bandwidth becomes broader, and the entire frequency band of 0 to 20 KHz is divided into, e.g., 25 critical bands.

An allowed noise calculation circuit 20 calculates allowed noise quantities every respective critical bands in which the so-called masking effect, etc. is taken into consideration on the basis of spectrum data divided every critical band to calculate allocated bit numbers every respective critical bands on the basis of the allowed noise quantities and energies or peak values, etc. every respective critical bands. In dependency upon bit numbers allocated every respective critical bands by the adaptive bit allocation encoder 18, respective spectrum data (or DCT coefficient data) are requantized. Data thus coded are taken out through output terminal 19.

Here, the above-mentioned allowed noise calculation circuit 20 is supplied with minimum audible levels every respective bands from a minimum audible curve generator 32. Each minimum audible level is compared with an allowed noise level in which the above-mentioned masking effect is taken into consideration at a comparator 35. As a result, when the minimum audible level is higher than the allowed noise level, this minimum audible level is regarded as an allowed noise level. At this time, an approach is employed to divide the critical band into smaller regions by taking into consideration an error of the minimum audible level

particularly in bands where the critical bandwidth is broad to allow minimum audible levels every these small divided bands to be respective allowed noise levels, thus to carry out bit allocation every respective small divided bands.

The operation thereof will now be described with reference to Figs. 3 and 4.

Fig. 3 is a flowchart for explaining the operation, and Fig. 4 shows the example where one critical band B is divided into smaller regions BB (four regions in the example of Fig. 4).

Initially, a step S1 of Fig. 3, there is carried out discrimination as to whether or not the level of the minimum audible curve RC of the small band BB_1 on the lowest frequency side of the four small bands BB_1 to BB_4 of one critical band B is higher than the level of a masking spectrum which is a present allowed noise determined in consideration of the masking (RC>MS). When the discriminated result at this step S1 is YES (the level of the minimum audible curve RC is higher than the level of the masking spectrum MS), the operation proceeds to step S2 to consider the allowed noise as a minimum audible curve RC to raise or set a flag F_{RC} at the next step S3 (F_{RC}=1). Subsequently, the operation proceeds to step S4 to conduct an adaptive bit allocation in dependency upon the level of the minimum audible curve RC which is an allowed noise to carry out coding. On the contrary, when the discriminated result at the step S1 is NO, the operation proceeds to step S5 to consider the allowed noise as

a masking spectrum to set the flag F_{RC} to 0 at step S6 to proceed to the above-mentioned step S4.

Here, when attention is drawn to one critical band B as shown in Fig. 4, the case where the minimum audible curve is RCa with respect to the masking spectrum MS as an allowed noise obtained at present corresponds to the case where the discriminated result at the step S1 is YES, and the case where the minimum audible curve is RCb or RCc corresponds to the case where the discriminated result at the step S1 is NO. When the minimum audible curve is RCa, this minimum audible curve RCa becomes an allowed noise. Thus, bit allocation is carried out every small bands BB_1 to BB_4 in dependency upon allowed noise levels given every small bands BB_1 to BB_4. On the contrary, when the minimum audible curve is RCb or RCc, the allowed noise becomes the masking spectrum MS. Thus, bit allocation is carried out in dependency upon a single allowed noise level in the critical band B.

Meanwhile, in the case of transmitting the allowed noise level as auxiliary information along with quantized main information, even when the minimum audible curve RCa is considered as an allowed noise, information transmitted is only a single allowed noise level in the critical band. This is because since the minimum audible curve is determined from the hearing sense characteristic of the human being, a minimum audible curve pattern or relative value data, etc. is caused to

be stored in advance into a ROM, etc., thereby making it possible to easily determine the minimum audible level of other small bands BB_2 to BB_4 on the basis of the minimum audible level of, e.g., the small band BB_1.

Fig. 5 is a flowchart for explaining the essential part of the decoding processing on a decoder side. At step S11 of Fig. 5, discrimination as to whether or not the flag F_{RC} is 1 is made. When the discriminated result is YES, i.e., an allowed noise of the corresponding critical band is given by the minimum audible curve, allowed noise levels every respective small bands BB_1 to BB_4 are calculated at the next step S12. Namely, even if only one allowed noise level with respect to a single critical band B, e.g., an allowed noise level NL_1 of the small band BB_1 on the lowest frequency side is sent as shown in Fig. 6, allowed noise levels NL_2 to NL_4 every respective small bands BB_2 to BB_4 can be determined by calculation from the pattern of the minimum audible curve RC by making use of a relative list, etc. of minimum audible values stored in a ROM, etc. as described above.

Further, when the discriminated result at the step S11 is NO, i.e., an allowed noise of the corresponding critical band is given by the masking spectrum MS, the operation proceeds to step S13 to set a fixed allowed noise in a single critical band B. On the basis of the allowed noise levels determined at these respective steps S12 and S13, bit allocation decoding processing is executed at the next step S14.

Meanwhile, the lengths of respective blocks (block sizes) in forming blocks in the time base direction every respective frequency bands divided by the above-mentioned division filters 11 and 12 to carry out floating processing thereafter to conduct orthogonal transform processing are adaptively switched in dependency upon an input signal.

Namely, explanation will be given in connection with the case where block size switching between a block BL having a large time width T_{BL} and blocks BL_{R1} and BL_{R2} each having a block length of $T_{BL}/2$ one half of T_{BL} as shown in Fig. 7. First, maximum absolute values MX_{R1} and MX_{R2} in respective blocks with respect to smaller blocks BL_{R1} and BL_{R2} are determined. Then, comparison between these maximum absolute values MX_{R1} and MX_{R2} is made. When the ratio therebetween is as indicated by the following equation (1);

$$MX_{R2}/MX_{R1} \geqq 20 \quad \cdots (1)$$

switching to the size of smaller blocks BL_{R1} and BL_{R2} is made. When otherwise, the size of the larger block BL is selected.

Then, explanation will be given in connection with the case where block size switching between a large block BL having a time width T_{BL}, medium blocks BL_{R1} and BL_{R2} having a block length $T_{BL}/2$ one half thereof, and small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4} having a block length $T_{BL}/4$ one half thereof. First, respective maximum absolute values MX_{S1}, MX_{S2}, MX_{S3} and MX_{S4} in blocks of the small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4} are determined. With respect to

these four maximum absolute values MX_{S1}, MX_{S2}, MX_{S3} and MX_{S4}, when the following relationship as indicated by the following equation (2) holds;

$$MX_{Sn+1}/MX_{Sn} \geqq 20 \quad \cdots (2)$$

where n is 1, 2 or 3

the block size of the small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4} having a length of $T_{BL}/4$ is selected. In contrast, when the above equation (2) is not satisfied, respective maximum absolute values MX_{R1} and MX_{R2} in blocks of the medium blocks BL_{R1} and BL_{R2}. Whether or not the following equation (3) is satisfied is discriminated.

$$MX_{R2}/MX_{R1} \geqq 10 \quad \cdots (3)$$

When the above equation (3) is satisfied, the block size of the medium blocks BL_{R1} and BL_{R2} having a length $T_{BL}/2$ is selected. On the other hand, when otherwise, i.e., the following equation (4) holds,

$$MX_{R2}/MX_{R1} < 10 \quad \cdots (4)$$

the block size of the large block BL having a length T_{BL} is selected.

Here, Fig. 9 shows the procedure in realizing by software the processing from data input of respective words of an input digital signal to the orthogonal transform processing. In Fig. 9, at step S111, absolute values of respective words are first calculated. At the next step S112, a maximum absolute value is detected. In place of detecting the maximum absolute value, a logical sum operation may be performed. At the next step S113,

detection of a maximum absolute value of all the words in one block or discrimination as to whether or not logical sum operation thereof is completed is carried out. This block is a block of respective selectable block sizes. When it is discriminated at the step S113 that the logical sum operation of all the words is not completed (NO), the operation returns to the step S111. In contrast, when the logical sum operation of all the words is completed (YES), the operation proceeds to the next step S114.

Here, in the case of taking a logical sum of absolute values in a block at the step S112, the processing for detecting a maximum absolute value in a block becomes unnecessary. Thus, floating coefficients (shift quantities) can be determined by a simple processing including only a logical sum operation.

The steps S114 and S115 correspond to the operation for detecting a shift quantity as a floating coefficient. At the step S114, left shift is carried out. At the step S115, whether or not the fact that a Most Significant Bit (MSB) of the shift result is equal to "1" is detected is discriminated. When "1" is not detected as MSB at the step S115 (NO), the operation returns to the step S114. In contrast, when "1" is detected (YES), the operation proceeds to the next step S116.

At the step S116, whether or not a maximum absolute value (or shift quantity) of all blocks of the respective sizes is obtained is discriminated. When the discriminated result is NO,

the operation returns to the step S111. In contrast, when the discriminated result is YES, the operation proceeds to the next step S117. At the step S117, a block size is determined on the basis of the above equation (1) or the above equations (2) to (4) to calculate a maximum absolute value of the block thus determined. At the next step S119, respective words are normalized (are subjected to floating processing). At step S120, whether or not all the words in the determined block are normalized is discriminated. When the discriminated result is NO, the operation returns to the step S119. In contrast, when the discriminated result is YES, the operation proceeds to the next step S121. At the step S121, discrimination is made as to whether or not, when, e.g., the block size of the medium blocks BL_{R1} and BL_{R2} or the small blocks BL_{S1}, BL_{S2}, BL_{S3} and BL_{S4}, etc is selected, the processing with respect to all blocks in the range of the large block BL is completed. When the discriminated result is NO, the operation returns to the step S111. In contrast, when the discriminated result is YES, the operation proceeds to the next step S122. At the step S122, the orthogonal transform processing is carried out. The processing is thus completed.

In accordance with this embodiment, by commonly using maximum absolute values (or logical sum outputs) calculated every respective blocks in determination of both the block floating coefficient and the block size, a quantity subject to processing

can be reduced. Thus, the number of steps, e.g., in the case of carrying out a processing by using the so-called microprogram can be reduced.

FIG. 10 is a circuit diagram showing, in a block form, the outline of the configuration of an actual example of the allowed noise calculation circuit 20. In FIG. 10, input terminal 21 is supplied with spectrum data on the frequency base from the respective DCT circuits 13, 14 and 15. As this data, an amplitude value of the amplitude value and a phase value calculated on the basis of a real number component and an imaginary number component of DCT coefficient data obtained as the result of execution of DCT operation is used. This approach is employed in consideration of the fact that the hearing sense of the human being is generally sensitive for the amplitude (level, intensity) on the frequency base, but is considerably dull for the phase.

Input data on the frequency base is sent to an energy calculation circuit 22 every frequency band, at which an energy every critical band is determined by using, e.g., a method of calculating sum total of respective amplitude values in a corresponding critical band, or any other method. In place of an energy every band, there are instances where a peak value, or a mean value of the amplitude value may be used. An output from the energy calculation circuit 22, e.g., a spectrum of sum total value of respective bands is generally called a bark spectrum.

FIG. 11 shows such a bark spectrum SB every respective critical bands. It is to be noted that, in order to simplify illustration in Fig. 11, the number of bands of the critical bands is represented by twelve bands (B_1 to B_{12}).

Here, in order to allow for the influence in the so-called masking of the bark spectrum SB, such a convolution processing is implemented to the bark spectrum SB to multiply it by a predetermined weighting function and to add the multiplied results. To realize this, an output from the energy calculation circuit 22 every band, i.e., respective values of the bark spectrum SB are sent to a convolution filter circuit 23. This convolution filter circuit 23 comprises, e.g., a plurality of delay elements for sequentially delaying input data, a plurality of multipliers (e.g., 25 multipliers corresponding to respective bands) for multiplying outputs from these delay elements by filter coefficients (weighting function), and a sum total adder for taking a sum total of respective multiplier outputs. By this convolution processing, sum total of the portion indicated by dotted lines in FIG. 11 is taken. It is to be noted that the above-mentioned masking refers to the phenomenon that a signal becomes inaudible as the result of the fact it is masked by another signal. As the masking effect, there are the time base masking effect by an audio signal on the time base and the simultaneous masking effect by an audio signal on the frequency base. Namely, by this masking effect, even if there is any noise

at the portion subject to masking, such a noise will be inaudible. For this reason, in an actual audio signal, noise within a range subject to masking is considered as an allowable noise.

Attention is now drawn to an actual example of multiplication coefficients (filter coefficients) of respective multipliers of the convolution filter circuit 23. Assuming that the coefficient of a multiplier M corresponding to an arbitrary band is 1, multiplying operation is carried out as follows: at the multiplier M-1, the filter coefficient 0.15 is multiplied by outputs from respective delay elements; at the multiplier M-2, the filter coefficient 0.0019 is multiplied by those outputs; at the multiplier M-3, the filter coefficient 0.0000086 is multiplied by those outputs; at the multiplier M+1, the filter coefficient 0.4 is multiplied by those outputs; at the multiplier M+2, the filter coefficient 0.06 is multiplied by those outputs; and at the multiplier M+3, the filter coefficient 0.007 is multiplied by those outputs. Thus, convolution processing of the bark spectrum SB is carried out. It is to be noted that M is an arbitrary integer of 1 to 25.

Thereafter, an output of the convolution filter circuit 23 is sent to a subtracter 24. This subtracter 24 serves to determine a level α corresponding to an allowable noise level which will be described later in the convoluted region. It is to be noted that the level α corresponding to the allowable noise

level (allowed noise level) is such a level to become in correspondence with the allowed noise level every band of the critical band by carrying out deconvolution processing as described later. Here, an allowed function (function representing the masking level) for determining the level α is delivered to the subtracter 24. By increasing or decreasing this allowed function, control of the level α is carried out. This allowed function is delivered from a (n-ai) function generator 25 which will be described later.

Namely, when the number given in order from a lower frequency band of bands of the critical band is assumed to be i, the level α corresponding to the allowed noise level is determined by the following equation:

$$\alpha = S - (n-ai) \quad \cdots (5)$$

where n and a are respectively constants (a>0), and S is intensity of a convolution processed bark spectrum. In the above equation (1), (n-ai) represents an allowed function. In this embodiment, n is set to 38 and a is set to 1. There results no degradation of sound quality at this time. Satisfactory coding is thus carried out.

In this way, the level α is determined. This data is transmitted to a divider 26. This divider 26 serves to apply deconvolution to the level α in the convoluted region. Accordingly, by carrying out this deconvolution, a masking spectrum is provided from the level α. Namely, this masking

spectrum becomes an allowed noise spectrum. It is to be noted that while the above-mentioned deconvolution processing requires complicated operation, simplified divider 26 is used in this embodiment to carry out deconvolution.

Then, the above-mentioned masking spectrum is transmitted to a subtracter 28 through a synthesis circuit 27. Here, the subtracter 28 is supplied with an output of the energy detector 22 every band, i.e., the previously described bark spectrum SB through a delay circuit 29. Accordingly, at this subtracter 28, a subtractive operation between the masking spectrum and the bark spectrum SB is carried out. Thus, as shown in FIG. 12, the portion of the bark spectrum SB of which level is lower than the level indicated by the level of the masking spectrum MS is subjected to masking.

An output from the subtracter 28 is taken out through an allowed noise corrector 30 and the output terminal 31, and is sent to a ROM, etc. (not shown) at which, e.g., allocated bit number information are stored. This ROM, etc. serves to output allocated bit number information every band in dependency upon an output (difference level between energy of each band and an output of the noise level setting means) obtained through the allowed noise corrector 30 from the subtracter 28. This allocated bit number information is sent to the adaptive bit allocation encoder 18, whereby spectrum data on the frequency base from the DCT circuits 13, 14 and 15 are quantized by bit

numbers allocated every respective bands.

Namely, in short, the adaptive bit allocation encoder 18 serves to quantize spectrum data every respective bands by bit numbers allocated in dependency upon levels of differences between energies of respective bands of the critical band and an output of the noise level setting means. It is to be noted that a delay circuit 29 is provided in order to delay the bark spectrum SB from the energy detector 22 by taking into consideration delay quantities at respective circuits preceding to the synthesis circuit 27.

Meanwhile, in synthesis at the above-described synthesis circuit 27, it is possible to synthesize data indicating the so-called minimum audible curve RC which is the hearing sense characteristic of the human being as shown in FIG. 13 delivered from a minimum audible curve generator 32 and the above-mentioned masking spectrum MS. In this minimum audible curve, if the noise absolute level is below the minimum audible curve, this noise cannot be heard. Furthermore, even if coding is the same, the minimum audible curve would vary, e.g., in dependency upon variation of a reproducing volume at the time of reproduction. However, it is to be noted that, since there is not so great variation in the manner in which a music enters, e.g., 16 bit dynamic range in actual digital systems, if it is assumed that quantization noise of, e.g., the frequency band most easily heard to ear in the vicinity of 4 KHz, quantization noise less than the

level of the minimum audible curve is considered to be not heard in other frequency bands. Accordingly, when a way of use in which noise, e.g., in the vicinity of 4 KHz of a word length that the system has is not heard is assumed to be employed, and an allowed noise level is provided by synthesizing the minimum audible curve RC and the masking spectrum MS, the allowed noise level in this case is permitted to be the level up to the portion indicated by slanting lines in FIG. 13. It is to be noted that, in this embodiment, the level of 4 KHz of the minimum audible curve is caused to be in correspondence with the minimum level corresponding to, e.g., 20 bits. In FIG. 13, signal spectrum SS is shown together.

It is to be noted that, as explained with reference to Figs. 3 to 6, in the critical band where the minimum audible curve is considered as an allowed noise, bit allocation every small bands obtained by dividing the critical band into smaller bands is carried out. Namely, at the comparator 35, the minimum audible curve from the minimum audible curve generator 32 and the masking spectrum MS from the divider 28 are compared with each other. The compared result is sent to the synthesis circuit 27, and is taken out as a flag F_{RC} from output terminal 36. For example, in the bands B_{11} and B_{12} of Fig. 13. since the level of the minimum audible curve RC is higher than the level of the masking spectrum MS, this minimum audible curve RC is regarded as an allowed noise, and the flag F_{RC} is set to 1. Thus, the

level of the minimum audible curve RC, e.g., on the lowest frequency side when the critical band is finely divided will be transmitted. As previously described above, calculation of allowed noise levels every respective bands is carried out on the decoder side.

At the allowed noise level corrector 30, the allowed noise level in an output from the subtracter 28 is corrected on the basis of information of, e.g., equi-loudness curve sent from a correction information output circuit 33. Here, the equi-loudness curve is a characteristic curve relating to the hearing sense characteristic of the human being, and is a curve formed by determining sound pressures of sound at respective frequencies which is heard at the same intensity of that of a pure sound, e.g., at 1 KHz to connect them. This equi-loudness curve is also called an equi-sensitivity curve of loudness. Further, this equi-loudness curve is substantially the same curve as the minimum audible curve RC shown in FIG. 13. In the equi-loudness curve, for example, in the vicinity of 4 KHz, sound is heard at the same intensity as that at 1 KHz even if the sound pressure is lowered by 8 to 10 dB as compared to that at 1 KHz. In contrast, in the vicinity of 50 KHz, sound cannot be heard at the same intensity as the at 1 KHz unless the sound pressure at 50 KHz is higher by about 15 dB than that at 1 KHz. For this reason, it is seen to allow a noise (allowed noise level) above the level of the minimum audible curve to have a frequency

characteristic given by a curve corresponding to the equi-loudness curve. From these facts, it is seen that it is adapted for the hearing sense characteristic of the human being to correct the allowed noise level by taking the above-mentioned equi-loudness curve into consideration.

Here, the correction information output circuit 33 may be of such a structure to correct the above-mentioned allowed noise level on the basis of information of an error between a detected output of output information quantity (data quantity) in quantization at the encoder 18 and a bit rate target value of the final coded data. The reason why such a correction is made is as follows. In general, there are instances where total number of bits obtained by applying, in advance, temporary adaptive bit allocation to all bit allocation unit blocks may have an error with respect to a fixed bit number (target value) determined by a bit rate of the final coded output data. In such instances, bit allocation is made for a second time so that the above-mentioned error becomes equal to zero. Namely, an approach is employed such that when the total allocated bit number is less than the target value, bit numbers of difference are allocated to respective unit blocks to add insufficient bits, while when the total allocated bit number is greater than the target value, bit numbers of difference are allocated to respective unit blocks to reduce surplus bits.

To carry out this, an error from the target value of the

total allocation bit number is detected. In dependency upon this error data, the correction information output circuit 33 outputs correction data for correcting respective allocated bit numbers. Here, in the case where the above-mentioned error data indicates insufficient bit number, an increased number of bits are used per unit block. Thus, consideration can be made in connection with the case where the data quantity is greater than the target value. In contrast, in the case where the above-mentioned error data is data indicating remainder of bit number, a lesser number of bits can be used per each unit block. Thus, consideration can be made in connection with the case the data quantity is less than the target value. Accordingly, from the correction information output circuit 33, in dependency upon this error data, data of the correction value for correcting an allowed noise level in an output from the subtracter 28, e.g., on the basis of information data of the equi-loudness curve is outputted. A correction value as described above is transmitted to the allowed noise corrector 30. Thus, the allowed noise level from the subtracter 28 is corrected.

Further, there may be employed a configuration such that the above-described synthesis processing for the minimum audible curve is not carried out. In this case, minimum audible curve generator 32 and synthesis circuit 27 become unnecessary, and an output from the subtracter 24 is subjected to deconvolution at the divider 26, and is transmitted immediately to the subtracter

28.

Further, as shown in Fig. 14, in the case of carrying out the block floating processing and the floating release processing before and after the Inverse Orthogonal Transform (IDCT, i.e., Inverse Discrete Cosine Transform) processing on a decoder side, an approach may be employed to take a logical sum of absolute values of respective words in a block, thereby making it possible to determine a floating coefficient.

In Fig. 14, input terminal 51 is supplied with coded data on the frequency base as obtained from the output terminal 19 of Fig. 1. This coded data is sent to an adaptive bit allocation decoder 52, at which it is subjected to decoding processing. Such data on the frequency base which have undergone adaptive bit allocation decoding processing are sent to a block floating processing circuit 56, at which floating processing every block is implemented thereto. Thereafter, the data thus processed respectively undergo, at Inverse Orthogonal Transform (IDCT, i.e., Inverse Discrete Cosine Transform circuits 53, 54 and 55 in the example of Fig. 14) processing opposite to the processing at the respective orthogonal transform circuits 13, 14 and 15 of Fig. 1. These outputs from the inverse orthogonal transform circuits 53, 54 and 55 are sent to a floating release circuit 57, at which floating release processing every block is carried out on the basis of floating information from the block floating processing circuit 56. Outputs of respective bands from the

floating release circuit 57 undergo, by using synthetic filters 58 and 59, processing opposite to the processing by the band division filters 11 and 12 of Fig. 1 so that outputs in respective bands are synthesized. The output thus synthesized is taken out from the output terminal 60.

It is to be noted that this invention is not limited to the above-described embodiment, but is applicable, e.g., not only to a signal processing apparatus for an audio signal but also to a signal processing apparatus for a digital speech signal or a digital video signal, etc.

As described above, the coding apparatus for digital signal of this invention is adapted to carry out a block floating processing of an input digital signal by a variable length block thereafter to implement an orthogonal transform processing thereto. In this coding apparatus, by determining the length of a variable length block and a floating coefficient of the block floating on the basis of the same index, it is possible to reduce a quantity subject to quantization or the number of steps of a program.

Further, in accordance with the coding apparatus for digital signal, when allowed noise levels every critical bands are determined by the minimum audible level, bit allocation is carried out by allowed noise levels every small bands obtained by further dividing the critical band to only transmit a flag indicating this, thus to avoid the necessity of sending allowed

noise levels every small bands. Accordingly, accurate allowed noise levels can be provided without increasing auxiliary information quantity. This leads to the fact that signal quantity can be improved without degrading bit compression efficiency. In addition, even if an absolute value of the minimum audible limit level is altered later, compatibility can be maintained.